BBC RD 1979/9

RESEARCH DEPARTMENT REPORT

An preliminary investigation into the measurement of time and frequency response of listening rooms and control cubicles

R. Walker, B.Sc.(Eng.)

Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION May 1979

BBC RD 1979/9
UDC 534.861.2

A PRELIMINARY INVESTIGATION INTO THE MEASUREMENT OF TIME AND FREQUENCY RESPONSE OF LISTENING ROOMS AND CONTROL CUBICLES

R. Walker, B.Sc. (Eng.)

Summary

The effects of 'early' reflections in a room on the perceived sound quality from loudspeakers are discussed, with particular reference to small rooms. Such reflections occur because surfaces within the rooms and the room boundaries themselves are not perfectly absorbing at all audio frequencies. In the majority of cases, for example listening rooms and sound control rooms (cubicles), most of the surface of the walls at least is covered with some form of sound absorbing treatment but, as this is invariably frequency selective to some extent, it only serves to further colour the perceived sound because the reflected sound is itself coloured.

One method of describing the overall acoustic of any room is by measuring the sound pressure level at a prescribed listening position as a function of both time and frequency. A number of techniques for obtaining this three-dimensional response are described and results presented for two rooms using one of these techniques.

It is shown that the techniques are relatively simple in principle and that meaningful results can be obtained down to a lower frequency limit which is equal to the required time resolution. However, the interpretation of the results so obtained is not apparent and it is likely that the variability in the detail of the results is larger than those features representing significant acoustic differences.

Issued under the authority of

Head of Research Department

Research Department, Engineering Division,
BRITISH BROADCASTING CORPORATION

May 1979

(PH-202)

A PRELIMINARY INVESTIGATION INTO THE MEASUREMENT OF TIME AND FREQUENCY RESPONSE OF LISTENING ROOMS AND CONTROL CUBICLES

Section	Title	Page
	Summary	Title Page
1.	Introduction	1
2.	Measurement techniques	2
	2.1 Measurements using tone-burst	2
	2.2 Measurements using other Gated Signals	4
3.	Measurements using pulse functions	5
4.	Measurements using swept sinusoidal waveforms	7
5.	Presentation of results	7
6.	Results	7
	6.1 Listening Room 1	7
	6.2 Measurements in the Acoustic Test Room	8
7.	Conclusions	9
8.	References	15
	Appendix: Recorded r.m.s. maximum of sampled sinusoidal waveforms	17

A PRELIMINARY INVESTIGATION INTO THE MEASUREMENT OF TIME AND FREQUENCY RESPONSE OF LISTENING ROOMS AND CONTROL CUBICLES

R. Walker, B.Sc.(Eng.)

1. Introduction

Many authors suggest that the early reflections in a room or a loudspeaker enclosure may affect the quality of the perceived sound and that the usual method of equalising loudspeakers to give a uniform, level steady-state on-axis amplitude-frequency response in a free-field environment completely ignores the effects of reflections of the sound from surfaces within the real listening environment.[1,2,3,4,5,6,7,8] These early reflections may be troublesome because the loudspeakers which are usually positioned at a not very great distance from the nearest solid objects of substantial size, for example, the walls of the room, have different directional properties at different frequencies. In addition, most loudspeakers are more directional at high frequencies than they are at low frequencies. The consequence of the reflections of the off-axis radiation by the nearby objects is to perturb the frequency response by interference between the direct and reflected sound. In Ref. 1, Harwood and Gilford showed that the steady-state frequency response of a loudspeaker, as measured at the listening position in a simulated sound control room, could be accurately predicted at low frequencies by a vector summation of the direct sound and the first few reflections. Fig. 1 is a reproduction of their Fig. 13 and shows the measured overall frequency response of their experimental arrangement, together with the theoretical response, calculated from the vector sum of the direct sound and the first three reflections. This measured response was judged to be a credible representation of the subjective effects noticed during listening tests using the same experimental arrangement.

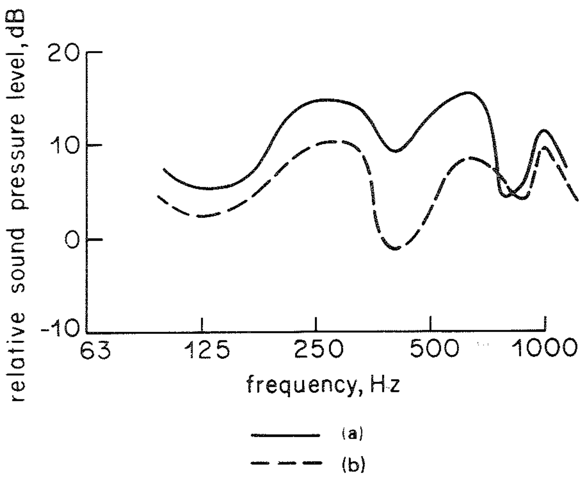

Fig. 1 - Amplitude/frequency characteristic of a loudspeaker in the corner of a room

(a) calculated from the direct sound plus the first three reflections
(b) measured (smoothed to remove minor irregularities)

This work was based on the assumption that the reflections themselves were not coloured by the wall treatment. In practice, acoustic treatment on the surfaces from which significant reflections are possible will modify the spectral distribution of the energy of the reflections and thus cause a further modification of the perceived steady-state response. In the extreme case of total absorption, the perceived frequency response will be the on-axis frequency response as measured under free-field conditions. In the other extreme case of very little absorption, the perceived frequency response will be that of the integrated total output power, as measured in a highly reverberant chamber.

In an effort to minimise the disturbance in the low-frequency response caused by these reflections, it is common practice that wherever possible, the surfaces near to and particularly those behind a monitoring loudspeaker are covered with an acoustic absorbing material which is effective at the lower frequencies, at which the loudspeaker is less directional. This technique significantly reduces the low-frequency reflections and thus makes the perceived frequency response much more like that measured under free-field conditions.

A second factor which is totally ignored in all such types of steady-state measurements is the psycho-acoustic effect of time-delay. Normal programme-type material has a near-gaussian amplitude probability distribution, at least at higher frequencies[5,9,10] and is thus much more like modulated noise than it is like a modulated sinusoid. In fact, the ear-brain relies on the transients in the sound waveform to provide direction information; with a normal stereo reproduction system it is almost impossible to locate the image of a source which has an uninterrupted sinusoidal waveform. Everyday experience indicates that when listening in a reverberant sound field, there is undoubtedly some limit of time-delay below which the ear-brain cannot distinguish between the echo and the direct sound. All reflected sounds which reach the ear before this time, relative to the direct

sound, are integrated with the direct sound in some way, to provide an overall impression of sound quality. Reflected sound which reaches the ear after this time will be heard either as reverberation if the reflection is diffuse or as an echo if the reflection is discrete. The effect is sometimes called the 'Haas effect' after Ref. 17.

A great deal of work has been done in an effort to determine the value of this critical time delay. References 6 and 11 to 22 represent but a small sample of the work which has been done. Values ranging from 5 ms (from Ref. 18 for transient sounds) to 90 ms have been claimed but it seems likely that a value of 50 ms is fairly representative.

The main purpose of this work was to investigate methods of measuring on a routine basis the three-dimensional response characteristics of rooms in which the reproduced sound quality is important such as listening rooms and control cubicles. The assumption was made that a correlation could be found between the objective and the subjective results. The characteristic to be measured was that of amplitude (sound pressure level) as a function of both frequency and time.

2. Measurement techniques

A number of methods are available for measuring the amplitude response as a function of both time and frequency. All of these methods can be grouped into three basic categories. First, using tone-burst or other gated measuring signals, second, using impulses as the test signal or third, by using a swept frequency source and offset narrow-band filter.

2.1 Measurements using tone-burst

In principle, this method consists of exciting the room with a loudspeaker driven by a short burst of the sinusoidal measuring frequency. The sound-field at the 'listening position' is monitored by an omnidirectional microphone and the resulting voltage waveform is processed by a 'gating' circuit. This gating circuit selects the wanted time-period and rejects the remainder of the waveform. A block diagram of the experimental arrangement is shown in Fig. 2 and a representation of the waveforms obtained is shown in Fig. 3. In practice, the

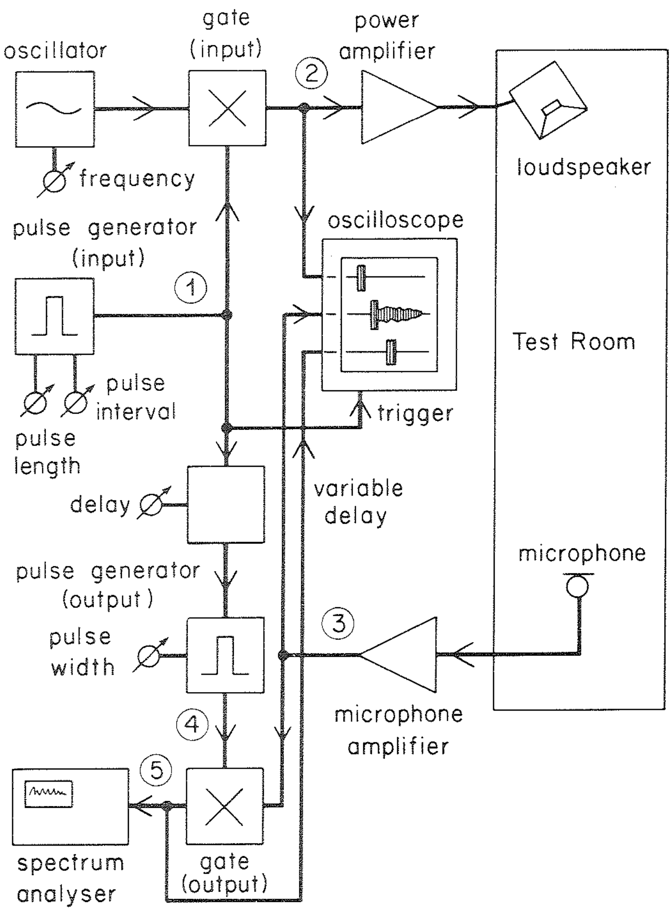

Fig. 2 - Block diagram of tone-burst apparatus (Numbers '⊗' refer to the points at which the waveforms are illustrated in Fig. 3)

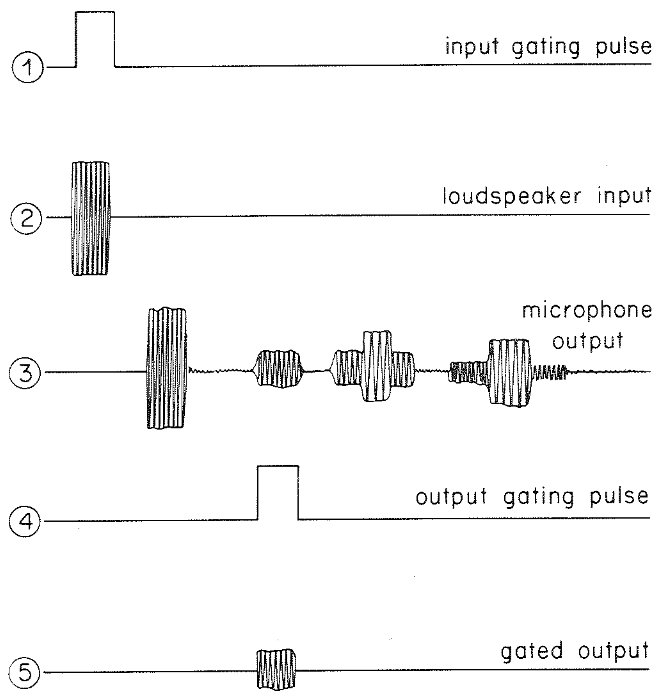

Fig. 3 - Representation of the waveforms obtained from the tone-pulse tests.

variable delay generator and the output pulse generator were implemented using the delayed time base facility of the oscilloscope. A simple interface between the 'delayed sweep gate output' of the oscilloscope and the output gate was all that was required. In this way, only a two-channel oscilloscope was needed, the selected portion of the microphone signal being indicated by the bright-up of the trace. The output from the gating circuit can be measured by any convenient means. In the experimental tests, the measuring instrument was a real-time 1/3rd octave band spectrum analyser set to record the maximum r.m.s. value obtained* in each frequency band. Thus, by setting the gating circuit to any particular delay and pulse width and sweeping the frequency slowly from one extreme of the required range to the other, the frequency response corresponding to a fixed time-delay was recorded. This procedure was repeated for a sufficiently large number of settings of the time-delay to cover the required temporal range. The ensemble of results so obtained is the three-dimensional amplitude/time frequency response; it has a temporal resolution equal to the gating pulse length and a frequency resolution proportional to the reciprocal of the gating pulse length.

One instrumental difficulty was the non-uniform frequency response of the exciting loudspeaker and receiving microphone combination. This was easily overcome by setting the time-delay for the first frequency response measurement to be equal to the time-delay for the direct sound. This effectively measured the free-field frequency response of the loudspeaker and microphone combination. This response was then subtracted from subsequent measurements so that the response relative to the direct sound was obtained.

A major fundamental difficulty, analogous to the Heisenburg uncertainty principle in quantum mechanics, was the theoretical interchangeability of time and frequency resolution. By using a vanishingly short pulse duration, the time resolution could, theoretically, be increased to give any desired accuracy. However, modulating a continuous sinusoidal function ('carrier' frequency) by a rectangular pulse gives rise to other frequencies in addition to original sinusoidal function. The frequency spectrum of these other frequencies, or 'sidebands' is of the form illustrated by Fig. 4. This limits the frequency resolution of the measurement at higher carrier frequencies where the modulation index can be fairly low. More significantly, at lower carrier frequencies where the modulation

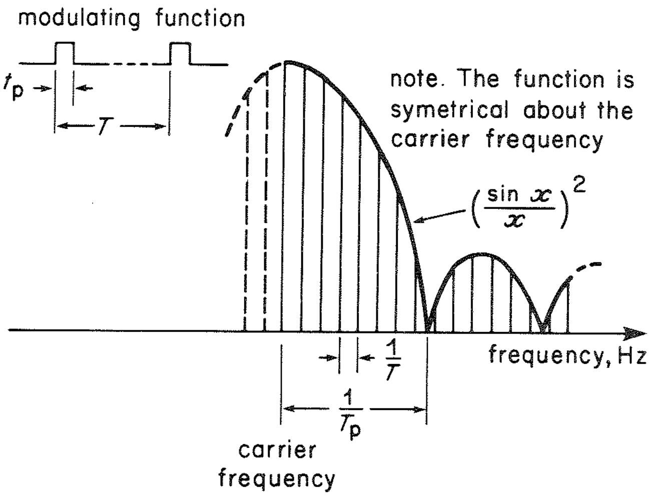

Fig. 4 - Frequencies generated by modulating a sinusoidal waveform amplitude by a rectangular pulse function.

index can be large, the measured amplitude of the carrier frequency can be less than that of the sidebands generated by a higher frequency. Thus the measurement of the response at low frequencies is 'swamped' by the sidebands generated by the measurement of the response at higher frequencies. Introducing tracking filters into different parts of the experimental arrangement would alter either the time resolution or the frequency resolution, depending on the position of the filter. For example, a tracking bandpass filter with a centre frequency equal to the carrier frequency and of suitable bandwidth could be positioned in the microphone circuit. The unwanted sidebands would then be rejected but the time resolution would be degraded as a result of the finite response time of the filter. Conversely, if the same filter were positioned after the sampling gate, then the time resolution would be that of the rectangular sampling function, but the frequency resolution would be degraded as a result of the sidebands generated by the sampling function itself.**

It was found that meaningful measurements could be made down to a frequency equal to the reciprocal of the pulse length, for example a 4 ms pulse length allowed meaningful measurements down to 250 Hz with a time resolution of 4 ms. This is approximately equivalent to a total path length difference of 1.4 m. For small rooms, in which the problems caused by early reflections seem to be most significant this performance is not adequate, particularly as the most serious effects

* See Appendix.

** It is interesting to consider that if this limitation is fundamental then it must also apply to the ear-brain combination and the perception of sound. Thus, it seems that these features cannot be of any significance and it is pointless to try to measure them.

generally occur at lower frequencies than 250 Hz.

An idea of the time resolution required to identify individual reflections in small rooms can be obtained by considering the experimental arrangements as set up in the Research Department Acoustic Test room. Fig. 5 shows the test layout which is typical of the situation in a moderate size sound control cubicle. Table 1 gives the calculated time of arrival at the microphone of the reflections from the various surfaces, relative to the direct sound. This table only includes reflections up to the second order. A reflection coefficient of only about 0.7 will give a sound pressure level of −6 dB relative to the direct sound (excluding the effect of distance) after two reflections. A sound reflection coefficient of at least 0.7 is quite likely unless the surfaces involved have been specially treated. Table 1 shows that, even for this not particularly small room, individual reflections can be very closely spaced, indeed there is no reason why some reflections should not be coincident in their time of arrival at the microphone under certain conditions.

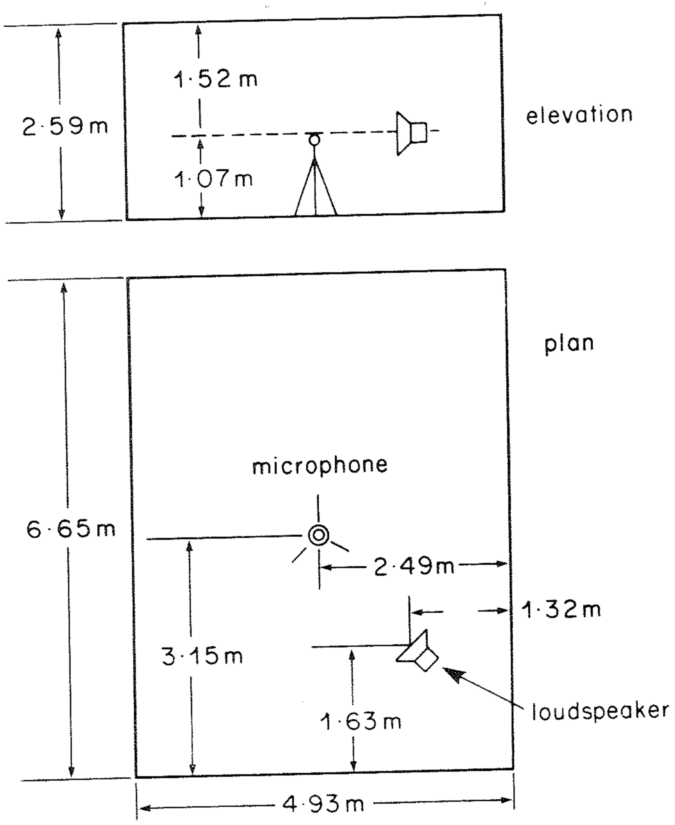

Fig. 5 - Experimental arrangement in the Acoustic Test room.

TABLE 1

Calculated time of arrival at the microphone of reflections, relative to the direct sound

Image	Time of Arrival ms	Path, via reflecting surface(s)
I_5	2.8	floor
I_6	5.0	ceiling
I_1	6.6	near side wall
$I_{1,5}$	8.8	near side wall, floor
I_2	8.9	front wall
$I_{1,6}$	9.5	near side wall, ceiling
$I_{2,5}$	10.2	front wall, floor
$I_{2,6}$	11.5	front wall, ceiling
$I_{1,2}$	12.5	near side wall, front wall
I_3	12.9	far side wall
$I_{3,5}$	13.9	far side wall, floor
$I_{3,6}$	14.9	far side wall, ceiling
I_4	20.0	rear wall
$I_{1,3}$	20.6	near side wall, far side wall
$I_{4,5}$	20.7	rear wall, floor
$I_{4,6}$	21.5	rear wall, ceiling
$I_{3,4}$	25.5	far side wall, rear wall
$I_{3,1}$	27.4	far side wall, near side wall
$I_{2,4}$	29.5	front wall, rear wall
$I_{4,2}$	38.6	rear wall, front wall

2.2 Measurements using other Gated Signals

Instead of using a tone burst, it is possible to use other pulsed test signals. In particular, a random signal which has equal energy per octave bandwidth ('pink' noise) is convenient as it allows measurements to be made at all frequencies simultaneously, when used on conjunction with a real-time spectrum analyser. The same limitations of frequency and time resolution apply as in the case of the pulsed sinusoidal test waveform.

3. Measurements using pulse functions

Instead of using a rectangular pulse to modulate a test waveform, as in Section 2 above, it is possible to use the pulse waveform itself as a test signal.[23] If the pulse is of sufficiently short duration it can be considered as an ideal impulse and the inverse Fourier Transformation can be used to calculate the frequency response.

However, the same theoretical limitation of the interchangeability of time and frequency resolution still applies. In order to obtain good time resolution, the overall impulse response of the system must be divided into short sections, each one representing the frequency response corresponding to its position in time. Fig. 6 shows a hypothetical impulse response of a particular loudspeaker, room and microphone combination. If a short section of duration τ, centred on a time delay of T is selected from this response, then the frequency response of the combination at time T can be obtained with a frequency resolution proportional to $1/\tau$ by carrying out the inverse Fourier Transformation of the short section selected, all of the remaining information being suppressed. However, this gating of the impulse response results in a modification of the signal and a reduction in the frequency resolution of the results. The effective output is given by the convolution of the actual frequency response with the inverse Fourier Transformation of the gating function.

The resolution can be improved by using some gating function other than a rectangular pulse, such as a Gaussian function, to select the wanted part of the impuse response. If the Gaussian weighting function has a half-amplitude duration of t seconds, then the wanted output is as if it had been analysed using a filter with a Gaussian frequency response with -3 dB points at approximately $\pm 1.8/3t$ Hertz relative to the centre frequency.

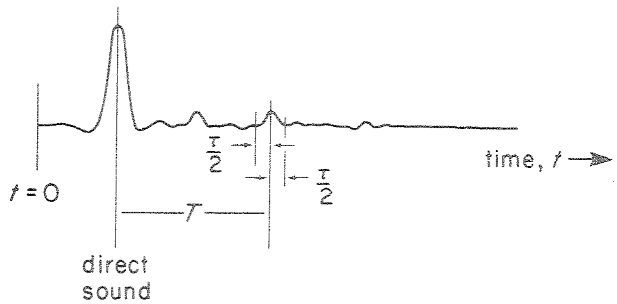

Fig. 6 - Hypothetical impulse response of loudspeaker, room and microphone combination.

Other gating functions, such as Hanning (cosine^2) or Hamming (cosine^2 + a small pedastal) can also be used. These two functions have the advantage of ease of generation, particularly in association with a Fourier transform, where the values of the cosine are required in any case in order to carry out the transformation. Both of these functions have finite sidelobe responses, but higher frequency resolution than the Gaussian function of the same total duration. Table 2 gives a summary of the relevant characteristics of these possible weighting functions, taken from Ref. 27.

TABLE 2

Function	-3 dB Bandwidth	Highest sidelobe response	Rate of decrease of sidelobe response
Rectangular	$\pm 0.9/\tau$	-13 dB	20 dB/decade
Hanning	$\pm 1.4/\tau$	-32 dB	60 dB/decade
Hamming	$\pm 1.3/\tau$	-42 dB*	20 dB/decade
Gaussian (truncated at $\pm 3.53\ \sigma$)	$\pm 1.8/3t$**	<-65dB	20 dB/decade

* Pedastal amplitude optimised for minimisation of the first sidelobe.

** The total duration of the Gaussian function has been taken as 3 times the half amplitude duration ($\tau = 3t$) that is, the function has been truncated at points corresponding to ± 3.53 standard deviations. This gives no sidelobe responses higher than -65 dB relative to the centre-frequency response. A true Gaussian function extends to infinity in both directions.

An alternative method of obtaining the impulse function is becoming much more viable with the increasing availability of fast, powerful computers. The method consists of exciting the room with a random or near-random function such as 'pink' or pseudo-random noise and using a computer to derive the autocorrelation function of the received signal.

This autocorrelation function is the same as the impulse function and thus restricted by the same limitations outlined above, just as if it had been measured directly.

A variation of the impulse technique which has been used to give a visual display of the total energy content of individual reflections is described in Ref. 24. A spark generator was used as the exciting function. This produced a sound pressure waveform of the form shown in Fig. 7, consisting

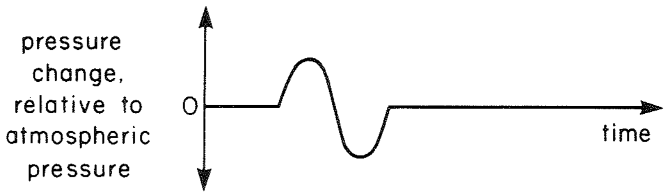

Fig. 7 - Approximation to the sound pressure waveform, used as the exciting function, from Ref. 24.

of a short pulse approximating to one complete cycle of a sinusoidal function. The resulting sound pressure waveform at any required position in the room being investigated was monitored by a suitable microphone and the resulting voltage waveform analysed. The equipment used to carry out the analysis first full-wave rectified the voltage waveform to produce a uni-directional signal, which was then integrated. The output of the integrator was reset to zero at a point corresponding to every other zero crossing of the input voltage waveform, the reset pulse being derived from the output of the full-wave rectifier by means of a slicing circuit and a divide-by-two circuit. Fig. 8 shows a block diagram of the apparatus and Fig. 9 a representative

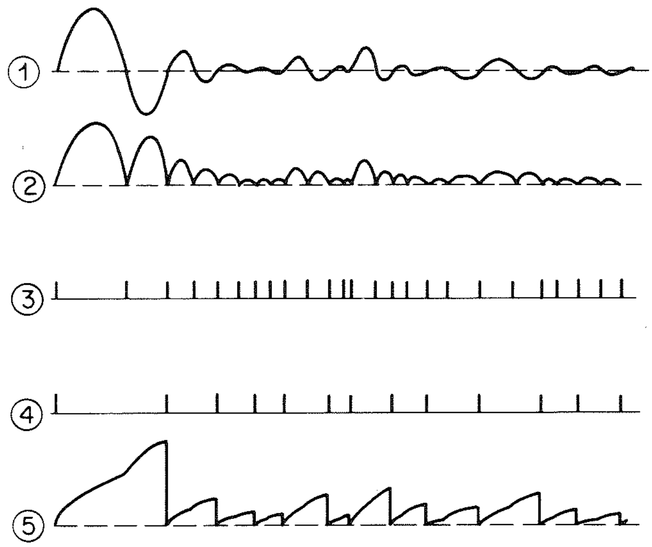

Fig. 9 - Typical waveforms produced by the apparatus of Fig. 8

selection from the various waveforms within the processing equipment. The result was a display of the integrated energy content of each reflection, provided that these reflections were spaced sufficient far enough apart in time for them not to

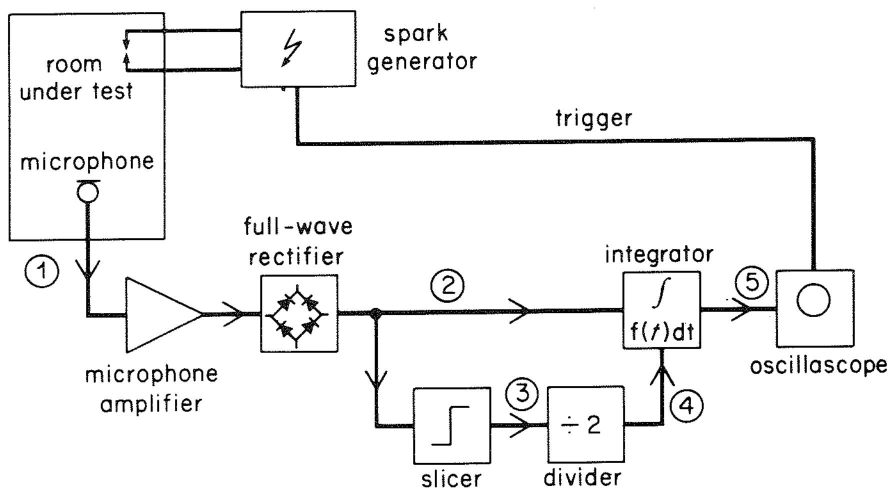

Fig. 8 - Block diagram of the apparatus used to display echoes directly (from Ref. 24). (Numbers 'x' refer to the points at which the waveforms are illustrated in Fig. 9)

overlap.

Once again, this technique is applicable to large rooms with distinct echoes and could be valuable for detecting unwanted reflections in concert halls or large studios, although it does not give any indication of the characteristic frequency response of the individual reflections.

4. Measurements using swept sinusoidal waveforms*

A fundamentally different method of measuring the three dimensional frequency response of a room is one which uses a continuous sinusoidal test waveform,[25] the frequency of which is swept at a constant (in Hz/second) rate from one end of the frequency range to the other. The microphone output is analysed by a narrow-band bandpass filter, the centre-frequency of which is offset from the frequency of the source by a fixed amount.

If the rate of sweep is S Hz per second and the filter frequency offset is Δf Hz, then the equivalent time delay is given by $\Delta f/S$ seconds. For example, if $S = 1$ kHz per second and Δf is 10 Hz, then the effective time delay is 10 ms.

This technique is limited in time resolution by the slow response time of narrow band filters and in frequency resolution by the finite bandwidth of the offset filter. The practical difficulties of providing the variable centre-frequency, fixed bandwidth offset filter are considerable, although there exists at least one piece of commercial equipment which will do this.

Reference 26 gives an account of one audio consultant who is currently able to offer this type of measurement, but makes no real attempt to interpret the results obtained.

Again, the compromise between time and frequency resolution, in this case evident in the choice of the bandwidth (and inevitably the response time) of the analysis filter, limits the resolution obtainable. A disadvantage of the method is that the frequency sweep must be linear in Hertz per second unless an exceedingly complex offset filter is used. This filter would have to maintain a fixed bandwidth whilst sweeping the centre frequency and the frequency offset in such a way as to match the oscillator at all times.

The use of a linear (in Hz per second) frequency sweep would result in a relatively long analysis time if the lower frequencies were to be measured to a high time resolution. For example, for a time resolution of 10 ms with a frequency offset of 10 Hz (i.e. a sweep rate of 1 kHz per second) the time taken to scan the entire spectrum up to 10 kHz would be 10 seconds.

* Based on an unpublished note by H.D. Harwood.

5. Presentation of results

Because of the limited time and equipment available, many of the techniques described in Sections 2 to 4 above could not be tested in practice. However, two series of measurements were made using gated tone burst (Section 2.1) and gated 'pink' noise (Section 2.2) respectively. One of the Research Department listening rooms and another room (ATR) were measured.

The experimental measurements were made in 1/3rd octave bands of frequency from 50 Hz to 10 kHz centre frequency (24 different frequency bands) for increments of 5 ms spacing in time over a range of 5 to 40 ms (8 different time ordinates). For each of these 192 points, a single figure representing the amplitude of the response, in dB relative to the direct sound, was obtained.

By the nature of the measurements, the results should ideally be presented in the form of a solid 3-dimensional model. One such model was actually constructed, Fig. 10, but it is evident that it is not possible to present this directly as a photograph or as a line drawing in two dimensions as so much of the detail is lost by being hidden from view. Because the measurements were quantised into 192 separate blocks, the model surface was discontinuous. For this reason a contour representation would have had no more meaning than the original two-dimensional array of amplitude measurments. In reality, the surface representing the response is not quantised but continuous. By interpolation, an approximation to this true surface could be reconstructed. It was then possible to draw a meaningful contour map to represent the three-dimensional surface in two dimensions. All of the results obtained are presented in this form.

6. Results

6.1 Listening Room 1

The experimental set-up using two loudspeakers in the listening room was arranged to be as nearly symmetrical as possible about the axis through the microphone (which was approximately in the position which would be occupied by a

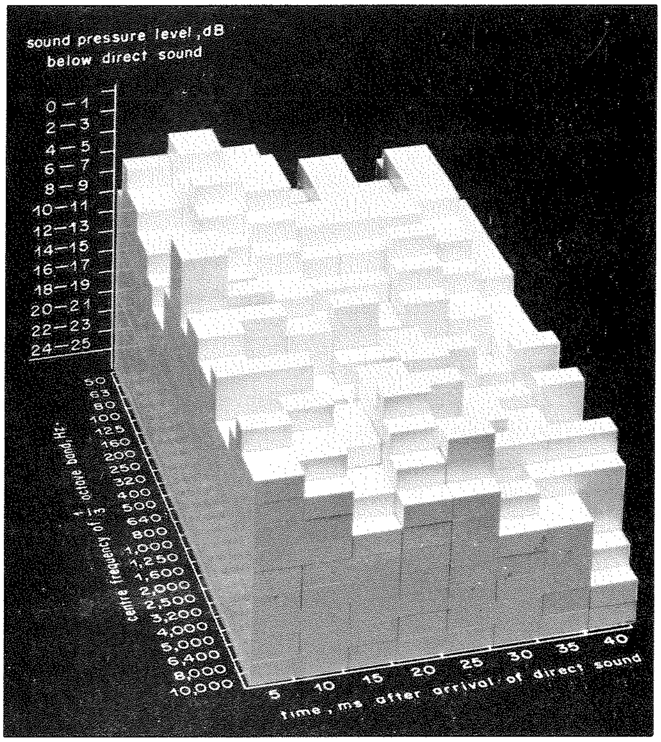

Fig. 10 - Model of 3-dimensional response of a room (ATR) as measured using gated 'pink' noise

listener) and the mid point between the loudspeakers. Subsequent measurements using phase cancellation of the direct sound from both loudspeakers simultaneously showed that the microphone was less than 15 mm off the axis of symmetry. The room itself was as nearly symmetrical as it reasonably possible in a real listening environment. Fig. 11 shows a sketch of the experimental arrangement.

Two measurements were then made of the time-frequency response of the room using the loudspeakers alternately and using the gated tone-pulse technique outlined in Section 2.1. The purpose of these measurements was to see to what degree the supposed symmetry was evident in the results. The duration of the tone-burst and the width of the measuring gate were 4 ms. Figs. 12, and 13 show the contour maps of the room response obtained by the method outlined in Section 5, the contours being in units of dB below the direct sound. Because of the pulse length used to obtain these results (4 ms), the region representing frequencies lower than about 250 Hz has little meaning for the reasons discussed in Section 2.1. Such detail as is present in this region is probably spurious or, at best, some function of the higher frequency detail.

Although these maps give a complete representation of the room response, it is not immediately obvious from a comparison of the two that any significant degree of similarity exists.

By extracting the contours at 0, 5, 10 . . ., 25 dB the diagrams can be considerably simplified. Figs. 14 and 15 show these simplified contour maps and a comparison between these two maps show that there are some general similarities, and also some large differences.

6.2. Measurements in the Acoustic Test Room

A second set of measurements was carried out in a different environment for two reasons. First, to compare the results obtained from a tone-

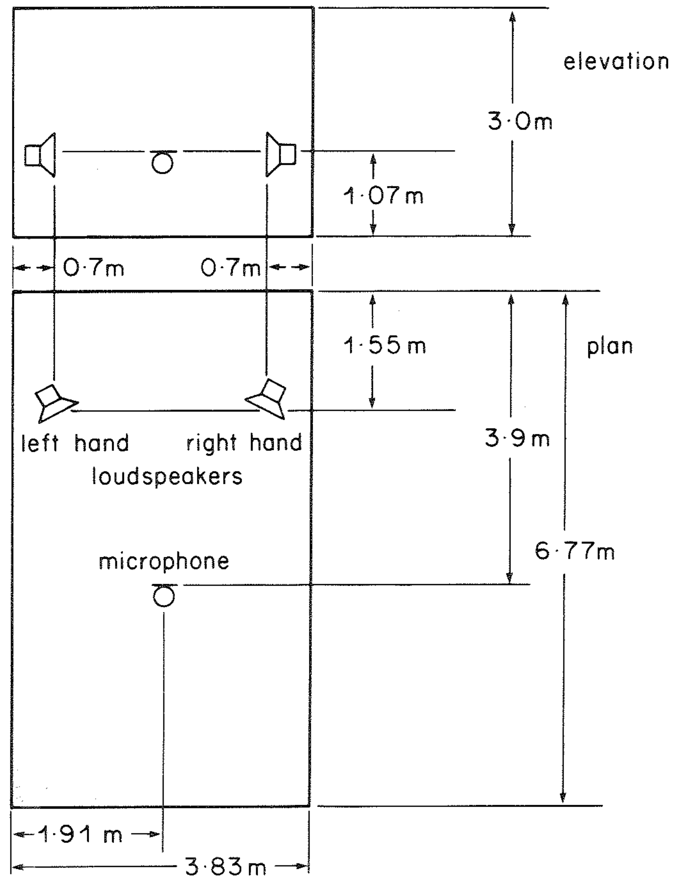

Fig. 11 - The experimental arrangement used for the measurements in Listening Room 1.

pulse test with those obtained by using gated 'pink' noise. Second, the acoustic test room was known to have acoustic anomalies, although it was treated with acoustic absorbers in much the same way as the listening room (although the actual units used were very different). Therefore, it was considered that it might be instructive to compare the results obtained in a room thought to be acoustically 'good' with those obtained in a room known to have some acoustic pecularities. Fig. 5 shows a sketch of the experimental set up. Fig. 18 shows the results obtained using tone burst. The same 4 ms pulse length was used as in the listening room. Comparison of Fig. 18 with Figs. 12 and 13 shows that at least as far as this type of measurement is concerned, no very great difference in character between the two rooms is very evident, except that the contour density is much higher in some parts of the middle-frequency range in the results for the ATR room than in the same areas in the results for the listening room. Comparison of Fig. 16 which is the simplified version of Fig. 18 with Figs. 14 and 15 shows that, for the ATR, there is a rather larger number of isolated peaks and troughs.

Finally, the results obtained from the gated 'pink' noise measurements are shown in Fig. 19 and the simplified version in Fig. 17. Comparisons between Figs. 19 and 18 and between Figs. 17 and 16 shows that there is a significant degree of similarity between the two sets of results obtained by tone burst and gated 'pink' noise at least at frequencies above 200 Hz. At frequencies lower than this, the results obtained using gated noise show much more detail than those obtained using tone burst. The reason for this increase in detail and whether or not it is real or false are not known at the present time, the theoretical limitations on frequency resolution should apply equally to both sets of measurements as stated in Section 2.2.

7. Conclusions

It has been shown that it is possible to measure the 3-dimensional time-frequency amplitude response of small rooms by a number of different methods. Two of these methods have been investigated experimentally in different rooms and the results compared. Although the method used to derive the presentation of the results involved a great deal of work in this case, it would be possible to programme a computer to greatly reduce the effort required. It would also be feasible to design a machine to automate the actual measurements. These two improvements would make it feasible to carry out this type of measurement on a routine basis.

However, the interpretation to be placed on the results is not evident at the present time, although it is possible that it might become evident if a much larger number of sets of results were obtained for different rooms. Even the two sets of results comparing the axial symmetry of a room which is generally thought to have no outstanding acoustic problems differed significantly from each other. It, therefore, seems likely that the variability in the detail of such measurements would be larger than that representing significant acoustic differences.

It has also been shown that it is theoretically impossible to obtain results of high temporal resolution at low frequencies. In fact, the lowest frequency which can be measured is equal to the reciprocal of the temporal resolution. This is a serious limitation of the method, particularly in small rooms, because to obtain any significant information about the room, the temporal resolution must be high, but the most serious acoustic problems in small rooms usually occur at low frequencies. However, this same limitation should apply equally to the hearing process and thus it is difficult to escape the conclusion that such prob-

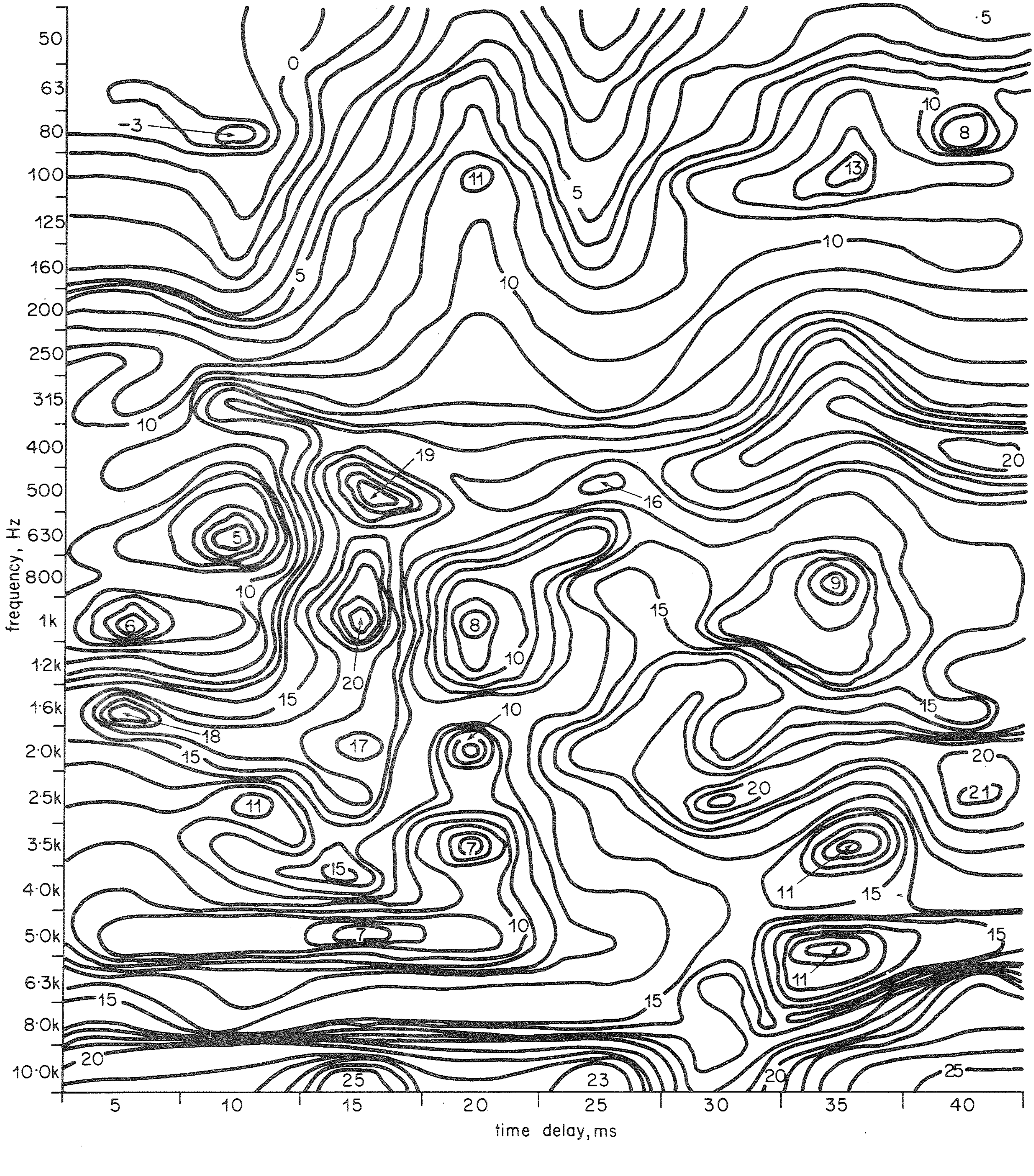

Fig. 12 - Contour map of the amplitude response as a function of time and frequency, in dB below direct sound (Listening Room 1, tone burst method, right-hand loudspeaker).

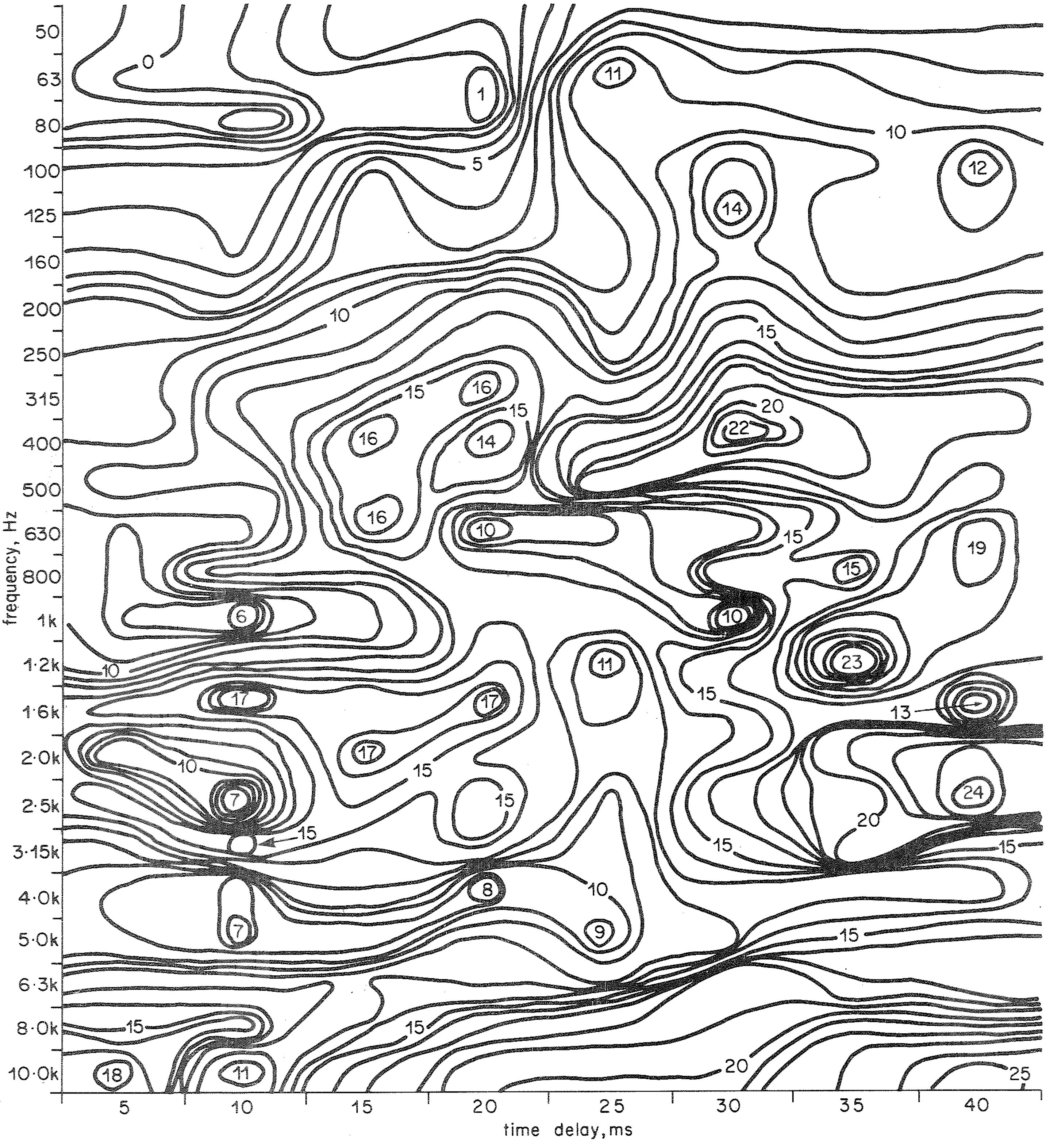

Fig. 13 - Contour map of the amplitude response as a function of time and frequency, in dB below direct sound. (Listening Room 1, tone-burst method, left-hand loudspeaker).

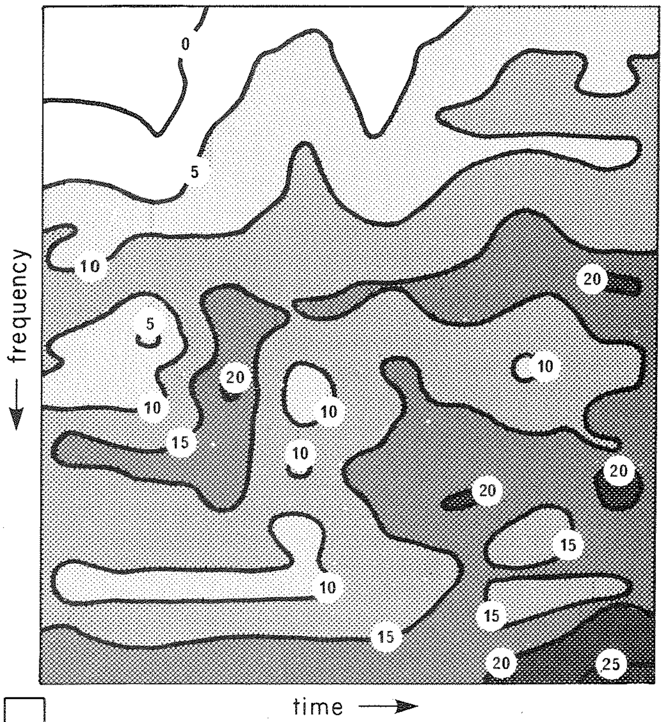

Fig. 14 - Simplified contour map of the amplitude response as a function of time and frequency, in dB below direct sound (Listening Room 1, tone-burst method, right-hand loudspeaker)

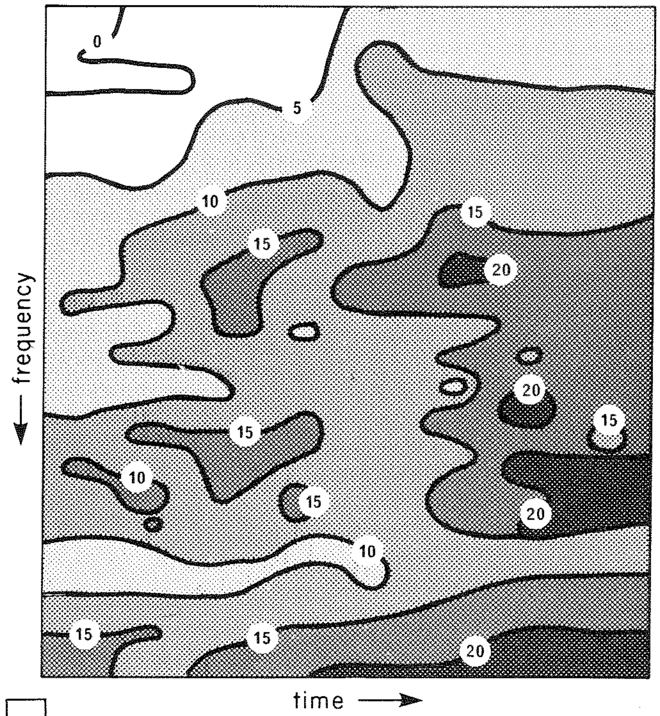

Fig. 15 - Simplified contour map of the amplitude response as a function of time and frequency, in dB below direct sound (Listening Room 1, tone-burst method, left-hand loudspeaker)

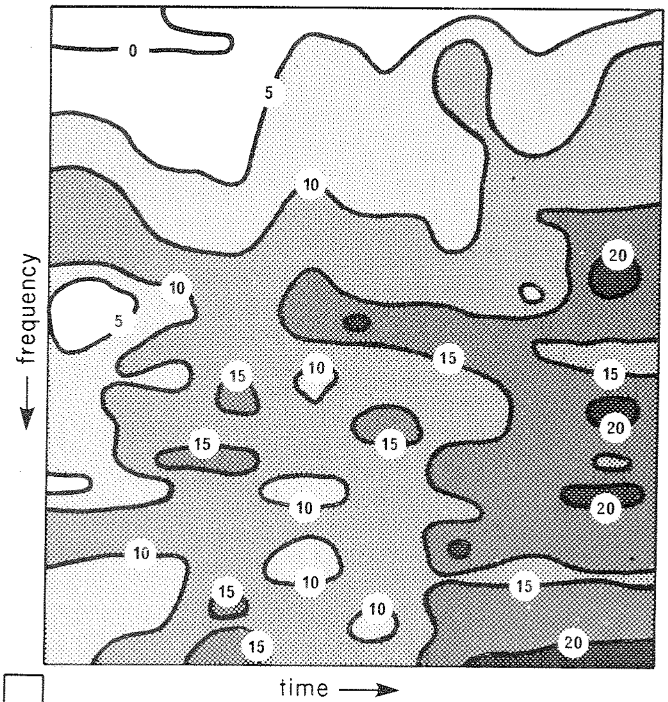

Fig. 16 - Simplified contour map of the amplitude response as a function of time and frequency, in dB below direct sound (Acoustic Test room, tone-burst method)

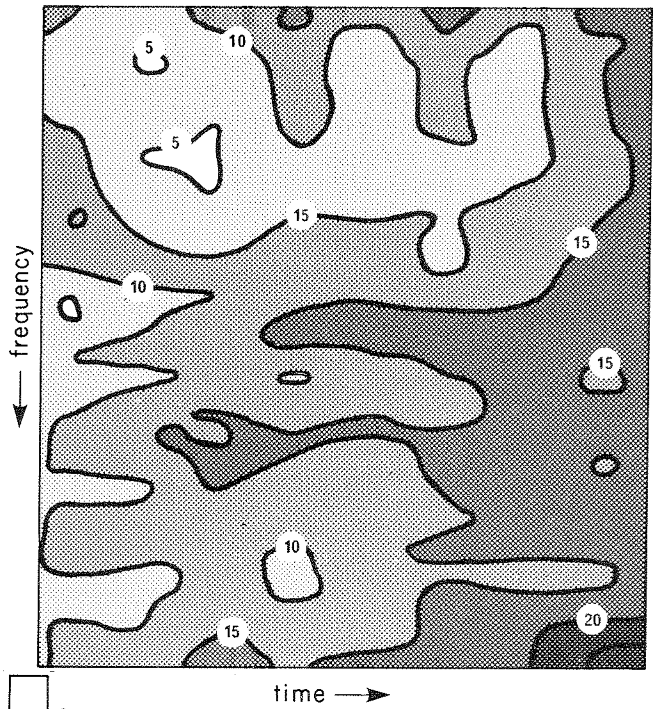

Fig. 17 - Simplified contour map of the amplitude response as a function of time and frequency, in dB below direct sound (Acoustic Test room, gated 'pink' noise method)

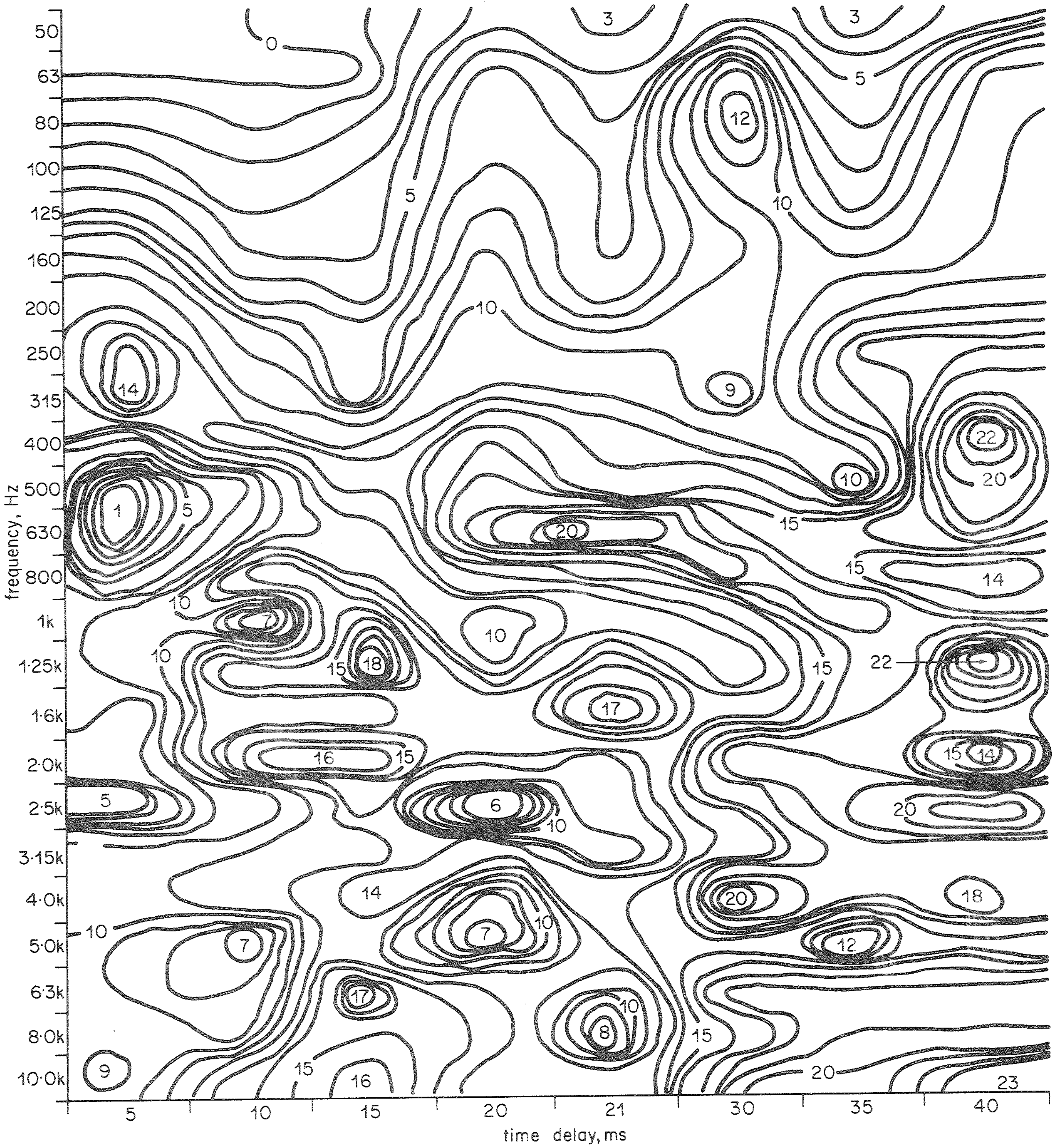

Fig. 18 - Contour map of the amplitude response as a function of time and frequency, in dB below direct sound. (Acoustic Test room, tone-burst method).

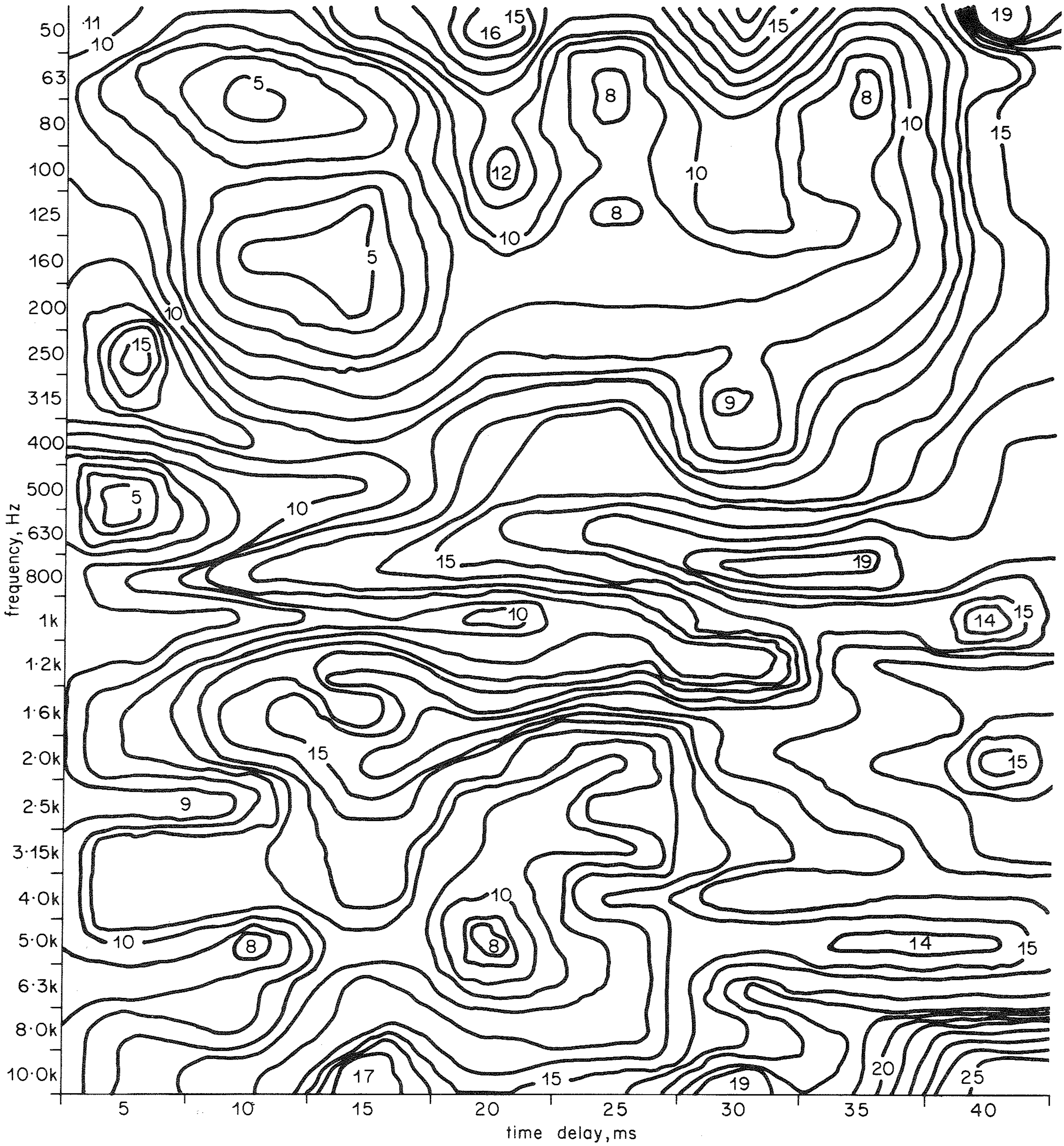

Fig. 19 - Contour map of the amplitude response as a function of time and frequency, in dB below direct sound (Acoustic Test room, gated 'pink' noise method).

lems are not connected with early reflections. A possible argument against this is that the ear-brain combination is an exceedingly complex signal processing chain, which may be able to use other clues about the environment, not measured by the simple apparatus described, to apparently exceed a theoretical limitation.

One of the assumptions made throughout this work was that the ear-brain integrates the sound at over a period of about 50 ms at all frequencies. If this period is not constant, but is a function of frequency or of the type of sound, then the range of time delays (and possibly the time resolution) used for these experimental measurements may be inappropriate. It would therefore be essential in any continuation of this work first to establish this integration time as a function of frequency and any other relevant variables. Having done this, valid results for the character of the room might be obtained simply by integrating the sound power incident at the microphone as a result of a short burst of tone or noise from the loudspeaker over whatever time period has been found to be appropriate.

8. References

1. HARWOOD, H.D. *and* GILFORD, C.L.S. 1969. Monitoring loudspeaker quality in television sound control rooms. BBC Engineering Division Monograph, No. 78, pp. 10 – 16.

2. QUEEN, D. 1974. Speaker performance. *Stereo Review,* August 1974, pp. 53 – 56.

3. ALLISON, R.F. 1974. The influence of room boundaries on loudspeaker power output. *JAES,* 1974, **22**, 5, pp. 314 – 319.

4. KROKSTAD, A. 1965. Loudspeaker in small rooms. 5th ICA, Liège, September 1965. Paper H55.

5. SCHULEIN, R.B. 1975. In situ measurements and equalisation of sound reproduction systems. *JAES,* 1975, **23**, 3, pp. 178 – 186.

6. QUEEN, D. 1971. Discrete field equalisation for sound reinforcement systems. *JAES,* 1971, **19**, 1, pp. 32 – 35.

7. QUEEN, D. 1975. Relative importance of the direct and reverberant fields to spectrum perception. *JAES,* 1973, **21**, 2, pp. 119 – 121.

8. ALLISON, R.F. *and* BERKOVITZ, R. 1972. The sound field in home listening rooms. *JAES,* 1972, **20**, 6, pp. 459 – 469.

9. SNOW, W.B. 1961. Loudspeaker testing in rooms. *JAES,* 1961, **9**, 1, p. 54.

10. SALAVA, T. 1971. Some aspects of the loudspeaker's performance evaluation in rooms. 7th ICA, Budapest, 1971, Paper 24 S 23.

11. QUEEN, D. 1973. Monitoring room acoustics. *dB.* May 1973, p. 24.

12. KORN, T.S. 1961. L'Effet de l'Entourage acoustique sur la résponse des haut parleurs. *Acustica,* 1961, **11**, 5, p. 323.

13. BROCINER, V. 1969. Loudspeakers – can we measure what we can hear. *Electronics World,* 1969, **81**, 3, p. 25.

14. OLSEN, H.F. 1957. Elements of Acoustic Engineering. Princeton, N.J., D. Van Nostrand Co. Inc., 1957, p. 575.

15. WEST, W. 1932. Acoustic Engineering. London, Pitman, 1932, p. 118.

16. MUNCEY, R.W. *and* NICHOLSON, A.F.B. 1964. The listener and room acoustics. *J S and Vib.,* 1964, **1**, 2, p. 141.

17. HAAS, Von. H. 1951. Über den Einfluss eines Einfachechos auf die Hörsamkeit von Sprache. *Acustica,* 1951, **1**, 2, p. 49.

18. WALLACH, H., NEWMAN, E.B. *and* ROSENZWEIG, M.R. 1973. The precedence effect in sound localisation. *JAES,* 1973, **21**, 10, p. 817.

19. MEYER, E. 1954. Definition and diffusion in rooms. *JASA,* 1954, **26**, 5, p. 630.

20. FUCHS, G.L. 1971. Integration time constant for annoyance of impulsive noise. 7th ICA, Budapest, 1971, Paper 24 S 11.

21. NICHOLSON, A.F.B., MUNCEY, R.W. *and* DUBOUT, P.D. 1954. The acceptability of artificial echoes with reverberant speech and music. *Acustica,* 1954, **4**, 4, p. 447.

22. ZWISLOCKI, J.J. 1969. Temporal summation of loudness; an analysis. *JASA,* 1969, **42**, 2 (Part 2), pp. 431 – 441.

23. CATERENOPOLOS, C.D. 1972. The measurement of the directivity characteristics of loudspeakers and microphones in reverberant enclosures. *IEEE Transactions on Audio and Electroacoustics,* 1972, **AU20**, 2, p. 151.

24. JUNIUS, W. 1958. Apparatur zur Automatischen auswertung von Schallrückwurffolgen. *Akustische Beihefte,* 1958, **8**, pp. S226 – S272.

25. HEYSER, R.C. 1967. Acoustical measurements by time delay spectrometry. *JAES,* 1967, **15**, 4, p. 370.

26. DAVIES, D. 1978. TDS in sound measurements. *Audio*, 1978, January, p. 46.

27. RANDALL, R.B. 1977. Application of Brüel & Kjaer equipment to frequency analysis. Brüel & Kjaer, 1977.

Appendix

Recorded r.m.s. maximum of sampled sinusoidal waveforms

If a sinusoidal waveform is sampled by a rectangular pulse, then the resultant r.m.s. voltage of the sampled output will vary depending on the phase of the sinusoidal function at the time corresponding to the beginning of the sampling interval. For repeatable results without a knowledge of this phase, the maximum value of the output function must be used.

Consider a unit sinusoidal function of frequency ω radians per second (f Hertz) sampled by a pulse of length t_p at intervals of T seconds, such that the beginning of the sampling interval occurs at a time equivalent to the angle θ of the sinusoidal function. This is illustrated in Fig. *A1*.

Duration of the sampling interval in terms of the sampled function

$$= f.t_p \text{ cycles}$$

$$= \omega.t_p \text{ radians}$$

Mean square voltage, MS, is given by:

$$MS = 1/T. \int_{t_1}^{t_2} (\sin^2 \omega t) \; dt$$

where $t_1 = \theta/\omega$

and $t_2 = t_p + \theta/\omega$

$$MS = 1/2T. \int_{\theta/\omega}^{t_p+\theta/\omega} (1 - \cos 2\omega t) \; dt$$

This gives

$$MS = (t_p - (\cos(2\theta + \omega t_p). \sin \omega t_p) /\omega) /2T$$

For any fixed frequency, ω, pulse length, t_p, and sampling interval T, the maximum value of this function occurs when

$$\cos(2\theta + \omega t_p) = -1$$

$$\text{that is } \theta = (\pi - \omega t_p)/2$$

and this maximum value is

$$\widehat{MS} = (t_p + (\sin \omega t_p) /\omega) /2T$$

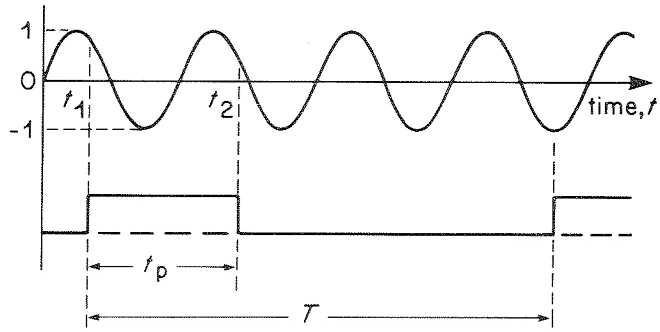

Fig. A1 - Unit sinusoidal function, sampled by rectangular pulses of duration t_p *at intervals of* T *seconds.*

But, ωt_p is the pulse length, expressed in radians of the sampled function.

$$\text{Let } \omega t_p = x$$

$$\widehat{MS} = t_p(1 + (\sin x)/x) /2T$$

Fig. *A2* shows this function.

If $x = n\pi$, where n is any positive integer, the term $(\sin x)/x = 0$ and $\widehat{MS} = t_p/2T$ which is also the value obtained for large values of x (i.e. the sampling inteval contains a large number of cycles.

For any fixed pulse length, the ordinate can be interpreted in terms of frequency. For example, for a pulse length of 4 ms, x_1 = 125 Hz, x_2 = 150 Hz, x_3 = 375 Hz, etc.

Two points should be considered in relation to Fig. *A2*. First, as stated in Section 2.1, meaningful measurements were not made at lower frequencies than that at which the sampling pulse length was equal to one complete cycle of the exciting frequency. This corresponds to the point $x = 2\pi$ in Fig. *A2*. For all higher frequencies the maximum error is approximately 1 dB. Second, as the same technique was used to measure the reflected and the direct sounds and the final result was expressed as the ratio between the reflected and the direct sound pressure levels, any such errors should cancel.

Thus it may be concluded that the correct parameter to measure is the maximum r.m.s. sound pressure level and the error arising as a result of the sampling (at frequencies higher than the lowest limit set by other considerations) is small.

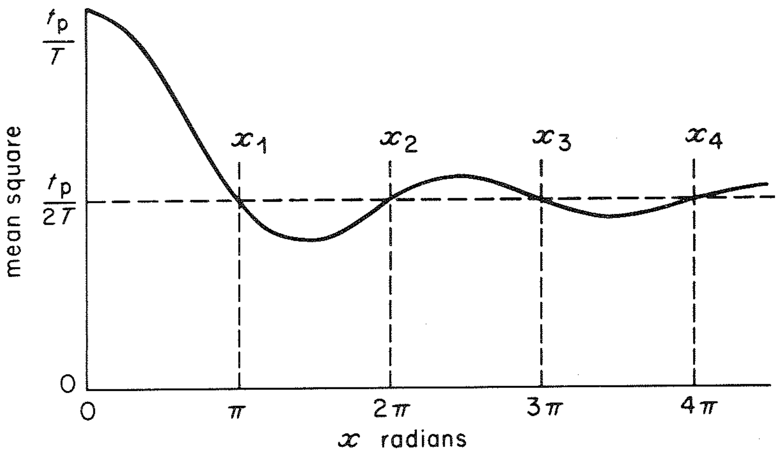

Fig. A2 - Maximum mean square output as a function of the sample pulse duration in radians.

CJS